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METHOD AND DEVICE FOR FORMATTING A DIGITAL AUDIO SIGNAL TO BE USED FOR SOUND REPRODUCTION

The principle for feeding electro-acoustic transducers by analog type amplifiers is most commonly used. In fact, the analog movement of the membrane follows sine waves which are curves sampled on recording. The condition for using the electric audio signal is the signal picked up by the micro-computer(s) or derived directly from digital samplers, said signal being smoothed for analog usage. The original digital sound signal used has an intensity and a frequency of 44,100 samples per second according to a well-known model (Fig. 1). The method is the original signal copied and reproduced to obtain a new digital signal for electrically controlling electro-acoustic transducers.

In fact, the pulse response of the electro-acoustic transducers does not conform to the signals emitted by the digital principle. A trailing effect can be observed on the pulses of the membranes which does do not stop when the pulse is finished. The mechanical pulse is extended by the effect of the weight of the membrane for strong pulses. The fine pulses are then masked in this particular condition. It becomes necessary to resolve this drawback by using the present method by imposing at least one additional item of information to control the mechanical effects of the runaway movements of the membranes.

The present method concerns an interface software containing a programmed special compensation control correction adapted for an electro-acoustic transducer.

Two possibilities have been retained to mitigate the mechanical runaway effects, either a multiplication of the digital audio signal with

smaller but in phase values, or phase inversion pulses. To achieve this, the method concerns using original reference digital sound signals which are copied and reproduced in several copies into a new disposition not modifying the time of the original. Control is precise in order to overcome the mechanical inertia of the membranes. The time parameter is a constant, also when the information doubles or triples the frequency of the signals double and triple proportionally to the multiplication of the reproduced copies. These copies of signals may have values of different intensities with respect to the original and with respect to one another. For example (Fig. 2), a time period is shown on the horizontal line where the values (EC) are present, the signals are copied and the time period does not change. The first copy reproduced has a scale ratio of 1/3 of the original and a second copy has a ratio of 1/1. It is clear that the multiple reproduction of the signals of the new sound signal corresponds to an adjustment of the frequencies of the doubled, tripled or quadrupled signals as many times as the original signal is multiplied. It is not echo which extends the note, but a method compensating the inertia of the membranes adapted to the dynamic stresses which by the weight dissociates the current voltage parameters of the instant. A smoothing of the newly formatted signal is provided to obtain an analog signal.

This digital or analog audio signal can be directly applied to the terminals of at least one electro-acoustic transducer.

This multiple reproduction of the original recorded signal may firstly have its intensity value different and secondly a phase shift, a sliding able to go up to its total inverse phase value with respect to the original signal. This method is characterised by the multiple reproduction of the originals (Fig. 3), known as the "original signal" with a given time (TP), into a new formatted signal 4 times the original frequency whose two copies are in inverse phase and with ½ intensity value. These phase inversion signals smooth-plane the membrane whose inertia would push it away too far and it would be late for the next pulse which creates current voltage dissociation.

The method can copy the original message and reproduce it according to all the variants of the combinations described identically for each original signal per order cyclic sequence reproduced in the intensity and phase scale ratio T1, T2, T3, T4 (Fig. 4).

The method has a device for formatting the audio signal for an electro-acoustic transducer, characterised by an original digital signal referential recorded on a medium or directly copied by reproduction of the original according to the method determining an interface software between an original digital signal and a newly formatted digital signal for an electric control of the best possible adapted mechanical movements. The new signals are at least two copies of the original signal. The copies of the signals may be in-phase or total in-phase inversion signals with respect to the original. The intensities of each signal can be a fraction of the intensity value of the original.

A device can be embodied by an expert in this field as an example (Fig. 4) in a sound reproduction chain having a radio receiver, a laser disk reader (CD) and a formatting system according to the method integrated in a digital amplifier (AN) for feeding the acoustic speakers (E). The amplifier receives the digital message via the optical beam (FO) and decodes the digital audio signal via the decoder (D) which shall establish the original signal. This signal is transferred to four samplers (T1, T2, T3, T4) managed by a common clock. A micro-computer (HC) fitted with a relay for regular and cyclic scanning programmes the activities of the four samplers able to be adjusted with respect to the values of the decoder (D). Each sampler, by copying the reproduced original signal, uses two programming potentiometers, namely one to position the scale ratio of the intensities, and the other for the phase of the intensities between the phase synchronicity and the phase inversion sliding with respect to the original. In this case, the sampler (T1) is identically adjusted as the sampler of the decoder of the original signal.

The sampler (T2) is adjusted to the inverse phase value of (T1) with the same intensity value. The sampler (T3) is adjusted to 1/3rd of the intensity of T1 and 1/2 the intensity of (T1). The sampler (T4) is in phase with (T1) and ½ of (T1). The formatted signal (F) is amplified by the amplifier (A) whose power in adjusted by the potentiometer (V) determining the sensitivity of the decoder.

The digital audio signal formatting devices according to the present method need a frequency 4 times greater for the medium and high notes and 3 times greater for the low notes. The sound reproduction system (Fig. 5), made up of a micro-computer, a digital audio cassette

recorder/reader and a laser disk reader and possibly other items are connected to the power digital amplifier with four independent outputs having 2 low notes (B) and 2 satellites (S). The amplifier requires decoders (RA) which establish the original signal. An electronic clock in an electronic chip (HE) co-ordinates the regular and cyclic programmed order of action of four samplers which determine the profile of the new sound message. The channels respectively receive three samplers coupled for the low notes (EGD, EGG) all phase-adjusted in a ratio of 1/3rd on reduction of the first with respect to the second and 1/3rd of the second with respect to the third. The signal newly formatted by the computer (X) is amplified by the amplifier (AX) so as to feed the low note boxes. The samplers (ESG, ESD) with four levels feed the satellites by means of the amplifiers (AX). The frequencies (ESG, ESD) have four times the speed of (RA), whereas those of (EGG, EGD) have three times the speed of (RA). A smoothing modem is provided for analog reading, this device not being restrictive and merely an example. All the electronic means with a semiconductors active or passive component or all the forms of micro-computers and integrated circuits or future products in the field of connections and active electronics can be used to embody these devices.

The present method and device adapts the digital sound signal into a digital signal for controlling all the electro-dynamic transducers. This correction principle can be used in audio and audiovisual applications.